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14. ABSTRACT
The objective of this effort was to adapt existing differential frequency hopping (DFH) algorithms developed for terrestrial communications for use in the doubly spread underwater acoustic channel by developing synchronization and demodulation schemes that are robust to various aspects of the environment using the DFH signal itself, and equalization algorithms that exploit available spatial diversity for improving the bit error rate (BER). The Sonar Simulation Toolkit (SST) was used to simulate the propagation of communication sequences through the underwater acoustic environment to specify an ocean environment with a wide variety of parameters relevant to acoustic signal propagation and reception. DFH data collected during at-sea experiments was used as a validation tool for simulated environments and more accurate modeling of underwater environments of interest for DFH-based acoustic communications.

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Differential Frequency Hopping (DFH) Modulation for Underwater Acoustic Communications and Networking

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Long Term Goals

The long-term goal of this research effort is to develop underwater acoustic communications algorithms based on differential frequency hopping (DFH) modulation that enable networked operations (i.e. multiple simultaneous users), as well as providing low probability of detection and intercept (LPD/LPI) and anti-jamming (AJ) capabilities.

Objectives

The specific objective of this effort is to adapt existing DFH algorithms developed for terrestrial communications for use in the doubly spread underwater acoustic channel. Key research goals are: (a) development of synchronization and demodulation schemes that are robust to various aspects of the environment using the DFH signal itself, and (b) development of equalization algorithms that exploit available spatial diversity for improving the bit error rate (BER). Variations in water depth, bottom type, sound speed profile, and source/receiver location can provide for wide ranges of multipath interference (resulting in time spread). Different wind and surface wave conditions in combination with platform motion can result in varying degrees of Doppler shift and spreading. The developed algorithms must work well across the range of conditions that might be encountered in real-life scenarios.

Approach

This project involves a close collaboration between Dr. Luca Cazzanti at APL/UW and Dr. Geoff Edelson at BAE Systems. Dr. Edelson has been involved in the development of the DFH algorithm [1], and has also been involved in a variety of other acoustic communications related projects for the Navy [2]. Dr. Cazzanti has experience in signal processing, adaptive equalization for digital communications, sonar, and underwater acoustics. Dr. Cazzanti also has experience with the Sonar Simulation Toolkit (SST), and guides the effort of Dr. Arindam Das, the APL-UW engineer primarily responsible for setting up, running, and analyzing the simulations needed for this project, and Mr. Chris Eggen, who is primarily responsible for designing the SST scenarios.

A key part of this effort is the use of SST [3] to simulate the propagation of communication sequences through the underwater acoustic environment. SST allows a user to specify an ocean

environment with a wide variety of parameters relevant to acoustic signal propagation and reception: sound speed profile, bathymetry, surface and bottom characteristics, ambient noise levels, etc. The user can also specify locations and trajectories of acoustic sources and receivers within that environment, and signals to be transmitted by the sources. SST then uses acoustic propagation models and time series simulation techniques to produce properly calibrated digital time series of the signals that would be "heard" by the receivers. These time series can then be operated on by signal processing algorithms, ideally the same algorithms that would operate on acoustic data measured at sea.

APL-UW provides the results of the SST simulation studies to BAE Systems, thus enabling algorithmic improvements by BAE to the DFH modulation schemes. The team has collaborated on the processing of the simulated signal, assessment of the algorithms' performance, and design of enhancements to improve performance for use in the underwater channel, with special regard to the effects of multipath and multiuser interference BER. Another key part of this effort is the analysis of DFH data collected during at-sea experiments. The measured data serves as a validation tool for our simulation environment and enables a better understanding and thus more accurate modeling of the underwater environments of interest for DFH-based acoustic communications.

Summary of work completed in prior years

In the first two years of this project, we investigated the performance of DFH in the underwater acoustic channel, in both SST-simulated environments and in sea-trial scenarios. We also showed that blind equalizers can work well with and are easily incorporated into the DFH receiver algorithm. First, we simulated several underwater ACOMMS scenarios using SST. The modeled scenarios included both soft- and hard-bottom environments to explore the effects of the ocean bottom on the multipath. We also simulated single- and multi-user scenarios. Then, we demodulated the SST-simulated received signals with a baseline DFH receiver and computed the BER to quantify the performance. Based on an analysis of these results, BAE Systems designed improvements to the baseline DFH receiver to mitigate multiple access interference (MAI). The improved DFH algorithm was then tested on data collected from the RACE08 sea trial and yielded very good multi-user performance.

We also investigated the applicability of blind equalizers to the DFH receivers to SST-simulated DFH signals. The results showed that the constant modulus algorithm (CMA) for blind equalization works well to mitigate intersymbol interference (ISI) in the received DFH signal. The important result was that off-the-shelf equalization strategies can be incorporated in the DFH receiver easily.

Finally, we characterized the environment of the RACE08 experiment in terms of SST model parameters, so that proof-of-concept simulations can be run before future experiments in similar environments.

Work completed

The major accomplishments during the third and final year of this project were:

- Refined the CMA blind equalizer for DFH communications and tested it on the RACE08 and SPACE08 experimental data.
- Refined the SST models by incorporating realistic environmental information from the sea trials.
- Published our findings in appropriate forums.

In the following sections we detail our work for the third and final year of research.

Blind equalization for DFH

In the previous two years of research, we investigated the use of equalizers to mitigate the effects of multipath-induced ISI on DFH modulation. Blind equalizers, that is equalizers that do not rely in a training sequence of bits, are a natural choice for DFH, which does not rely on training sequences for demodulation. Among blind equalizers, the constant modula algorithm (CMA) equalizer has been studied extensively and is well-known. Originally designed for terrestrial PSK communications, CMA equalization has also been shown effective for FSK-type modulations [4]. The CMA adaptively and iteratively trains the coefficients $\mathbf{c}$ of a finite impulse response (FIR) filter by gradient descent. The coefficients at iteration $k+1$ are computed with the update rule:

$$\mathbf{c}_{k+1} = \mathbf{c}_k - \mu E\left[\frac{\partial D}{\partial \mathbf{c}_k}\right]. \tag{1}$$

The residual D is the distance between the magnitude (or modulus) of a received symbol sequence Z_i and a data-dependent constant R_p,

$$D = \left(|Z_k|^p - R_p\right)^2, \; p = 2. \tag{2}$$

Applying CMA equalization to DFH modulation is straightforward. It is sufficient to insert the equalizer, which is a FIR filter, before the demodulator, as in Figure 1. Our preliminary results, based on SST simulations, showed that a CMA-based blind equalizer significantly reduced the multipath peaks in the channel impulse response for the difficult case of a highly reflective ocean bottom.

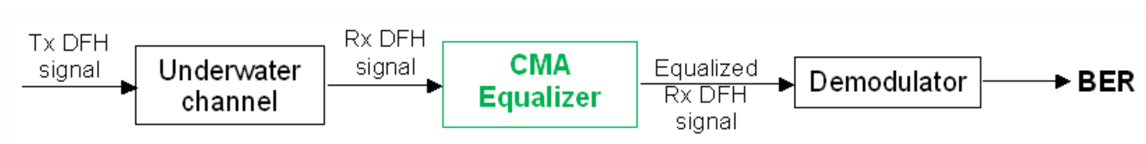


Figure 1: Signal processing with an equalizer module.

In this year's effort, we confirmed that CMA also performs well on real-life data collected during the RACE08 and SPACE08 experiments. As an example, Figure 2 shows the effect of equalization on a typical CIR from the RACE08 experiment. The CIR for the non-equalized case is computed by replica-correlating a transmitted linear FM chirp with its received version. Note the presence of secondary peaks in the response, due to multipath. These peaks cause ISI, which results in decoding errors at the receiver. For the equalized case, an equalizer was first trained on a single-user received DFH sequence (file (P04_T00_N02_R01_V01.wav). Then, the trained equalizer was used to filter the received linear FM chirp before replica correlation with its pristine transmitted version. Note that the secondary peaks in the CIR are attenuated due to the equalizer's effect. We quantified the performance improvement provided by the equalization by computing BERs for both the RACE08 and SPACE08 sea experiments. They are discussed in more detail in the following sections.

RACE08 equalization results

During the RACE08 sea trial, multiuser DFH sequences were transmitted in Narragansett Bay, Rhode Island from 1-17 March 2008. The corresponding received sequences were recorded and analyzed. The sound speed profile was approximately isovelocity, varying with the tides (primarily due to salinity changes) between 1455 and 1470 m/s. The surface conditions were primarily windblown chop. Much of the bottom type found in the Bay system is clayey-silt or sand-silt-clay [5], in either case, a relatively soft bottom.

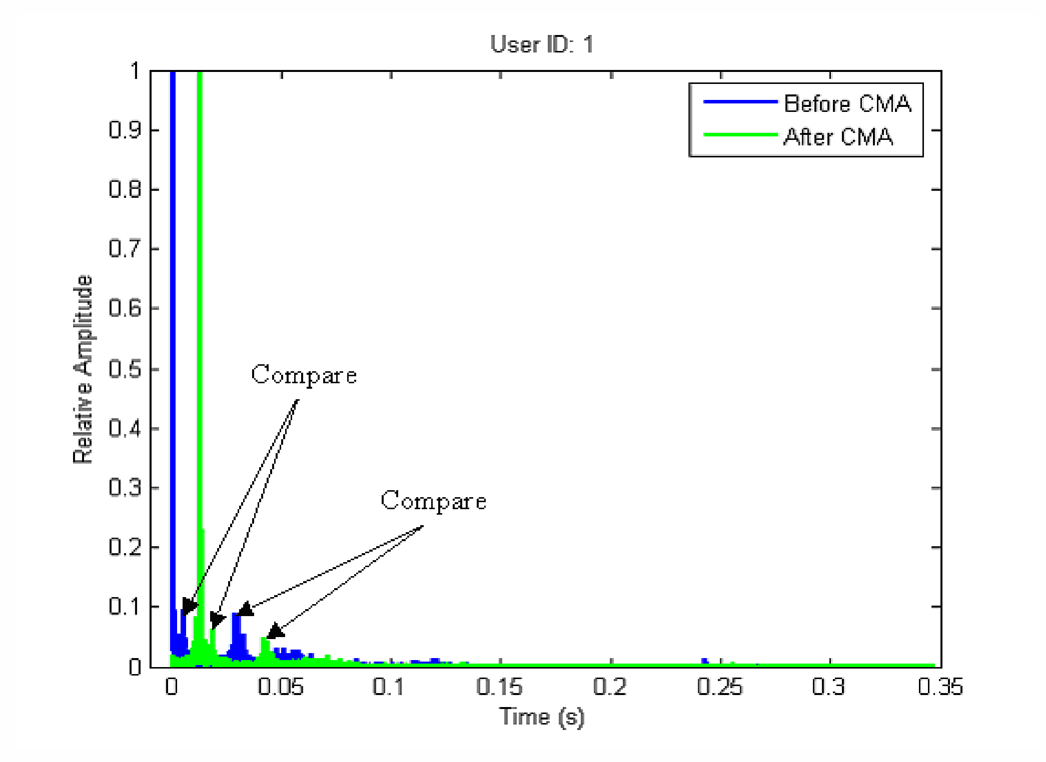


Figure 2: Comparison of normalized impulse responses a typical RACE08 reception. The equalizer attenuates the relative height of the secondary peaks, thus mitigating the interference due to multipath and improving the BER. The time offset is due to the delay introduced by the equalizer and does not affect the BER.

The experiment layout consisted of a four-element source vertical line array, a reference receiver element, and three receiver vertical line arrays as shown in Figure 3. The four transmitter elements were treated independently, as separate users. Linear Frequency Modulated (LFM) chirps were also transmitted with some signals to probe the underwater channel and obtain channel impulse responses (CIRs). These chirps were 1 second long and spanned the same bandwidth occupied by the DFH sequences (9-13 kHz), but are not utilized by the DFH modulation scheme: they are only used to obtain the channel impulse responses to characterize the environmental conditions. Note that although the experimental setup included receiver arrays, no array processing was done at all on the received DFH signals. Each receiver hydrophone was treated as an independent transmission, because the goal of our research was to characterize single-channel baseline performance for DFH.

We conducted an extensive evaluation of BERs on the experimental received signals, the results of which have been tabulated in Table 1. We report the BERs and the percent of error-free transmissions, which is a commonly-used metric for performance comparison. From Table 1, it can be seen that equalization leads to an improvement in average BER for almost all cases, except for user 1 when two users are transmitting simultaneously. However, even for this case, there is a substantial improvement in the median BER ($\approx 56\%$) due to equalization, which suggests the presence of a few outliers skewing the average BER metric. In fact, an analysis of received signals revealed that a few receptions contained non-DFH signals which were transmitted by mistake during the experiments, These few signals, for which the BER is 0.5, skew the overall results.

Figures 4(a) and 4(b), show comparisons of BERs for non-equalized and equalized signal receptions for user ID 4 when it is the sole transmitter and when all four users are transmitters. Data points above the diagonal represent an improvement due to equalization. Note that, consistent with the results from Table 1, equalization improves the BER. It is surprising that equalization

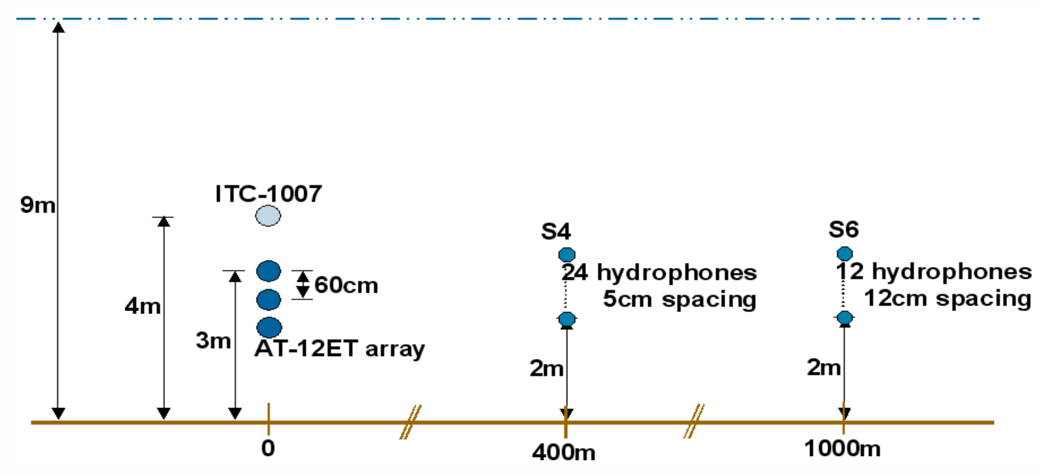
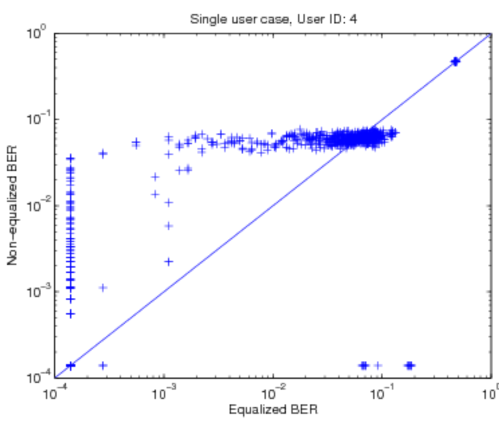
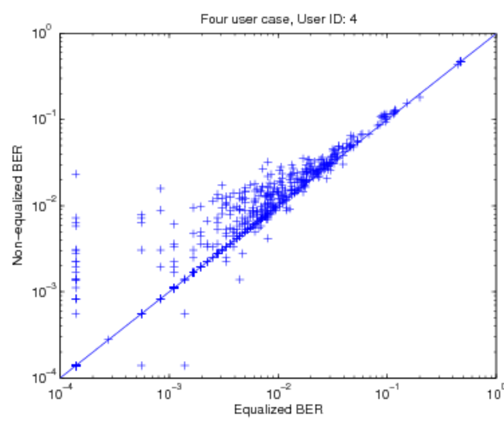


Figure 3: Array geometry for the RACE08 experiments.

improves the BER also for the case when multiple users are transmitted simultaneously. This is because the receiver first attempts to synchronize the reception to each of the four possible users, and then applies CMA equalization to each user separately. Effectively, each user's bit stream is synchronized and equalized individually while treating the other users as additive interference. For the RACE08 experiment, which exhibited benign environmental conditions with limited multipath, this strategy works well.



(a) Single user



(b) Four users

Figure 4: RACE08 experiments: comparison of BERs for several transmissions of user 4, when only one user is transmitted and when four users are simultaneously transmitted. Points above the diagonal indicate an improved BER due to equalization.

SPACE08 equalization results

The SPACE08 experiments were conducted at the Air-Sea Interaction Tower (ASIT), which is part of WHOI's Martha's Vineyard Coastal Observatory (MVCO), between October 13-28, 2008. The surface conditions during the trials were primarily wind blown chop and the sound speed profile was approximately isovelocity. The topography in the area is relatively benign with some transient small-scale features that are generated by passing storms. The bottom type at the experiment location is sandy. Following the passage of the first major storms of the season, the water column tends to stay well mixed for the winter resulting in a constant sound speed as a function of depth. The

Single User				
User ID	*Parameter*	*With eq.*	*Without eq.*	*% improvement*
4	Average BER Median BER % of error-free transmissions	0.024899 0.000139 63.23	0.023204 0.000139 68.80	6.8075 0
Two simultaneous users				
User ID	*Parameter*	*With eq.*	*Without eq.*	*% improvement*
1	Average BER Median BER % of error-free transmissions	0.035506 0.006401 26.80	0.038415 0.002783 34.13	− 8.1930 56.5224
2	Average BER Median BER % of error-free transmissions	0.019280 0.000139 51.51	0.013918 0.000139 59.93	27.8112 0
Three simultaneous users				
User ID	*Parameter*	*With eq.*	*Without eq.*	*% improvement*
1	Average BER Median BER % of error-free transmissions	0.135438 0.129975 11.64	0.131847 0.125522 11.64	2.6514 3.4260
2	Average BER Median BER % of error-free transmissions	0.035912 0.031450 0	0.031712 0.026997 0.19	11.6953 14.1590
3	Average BER Median BER % of error-free transmissions	0.045544 0.034233 3.23	0.042666 0.030476 3.04	6.3192 10.9748
Four simultaneous users				
User ID	*Parameter*	*With eq.*	*Without eq.*	*% improvement*
1	Average BER Median BER % of error-free transmissions	0.084898 0.047314 8.59	0.082202 0.042861 9.06	3.1751 9.4118
2	Average BER Median BER % of error-free transmissions	0.006508 0.003618 19.78	0.004509 0.002227 21.96	30.7165 38.4615
3	Average BER Median BER % of error-free transmissions	0.008771 0.002505 27.26	0.006724 0.001948 30.22	23.3362 22.2222
4	Average BER Median BER % of error-free transmissions	0.019431 0.007375 20.72	0.016879 0.005010 23.68	13.1354 32.0755

Table 1: RACE08 experiments: performance comparison with and without equalization.

dominant waves in the area tend to come from 240° True. At times during the experiment duration, winds reached high speeds, causing increased variability in the acoustic channel characteristics.

Five acoustic transducers were available for transmitting multiuser DFH sequences and channel probes, as shown in Figure 5. One transducer (transducer number 0) was located approximately 4 meters above the bottom on a stationary tripod. The remaining four transducers were deployed in a vertical array (numbered 1 through 4 from top to bottom) with the top transducer in the array approximately 3 meters above the bottom. The center-to-center spacing of the transducers in this array was 50 cm. Fixed receiving arrays were located on two paths southeast and southwest of the source array at ranges of approximately 80, 200, and 1000 meters from the source. The

arrays at 80 meters range were of a cross shaped configuration with 16 elements on each leg with a 3.75 cm spacing between elements. The arrays at 200 meters range were 24 element vertical arrays with a spacing of 5 cm between elements and the arrays at 1000 meters range were 12 element vertical arrays with 12 cm spacing between elements. Finally, a reference hydrophone was mounted approximately one meter from transducer number 0. Note that we did not perform any array processing on the received DFH signals, treating instead each hydrophone as an independent receiver.

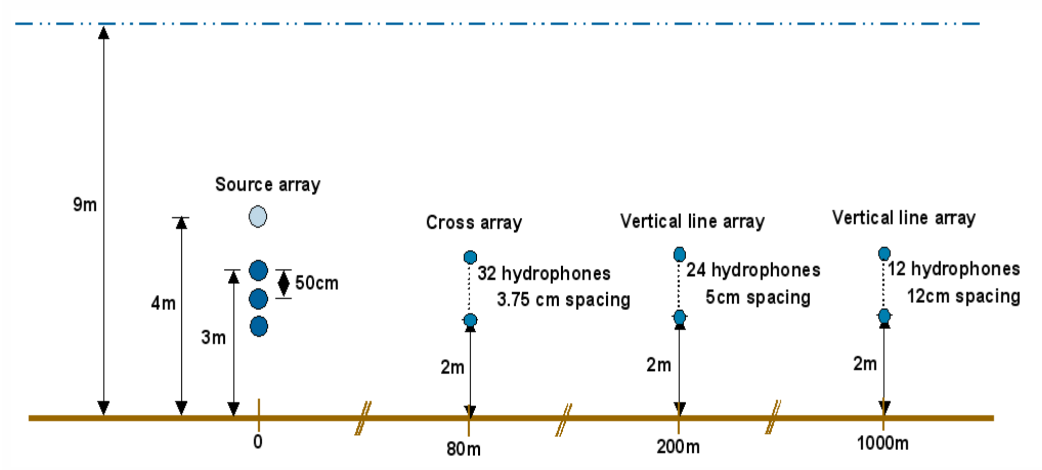


Figure 5: Array geometry for SPACE08 experiments.

Table 2 shows the results of applying equalization to the SPACE08 experiments. Equalization improves both the average and median BERs, even for the case of of multiple users. As the number of simultaneous users increases, MAI becomes the dominant effect over multipath, and the benefits of equalization are diminished. Future research must address MAI in scenarios similar to the SPACE08 experiment, where tougher environmental conditions make it more difficult for the receiver to synchronize to each user's transmission and thus limit the benefit provided by equalization.

Refining the SST models

One of the original goals of this research was to develop SST models to simulate the transmission of DFH waveforms in the underwater acoustic channel. During the previous two years of research we developed a notional SST scenario for quick proof-of-concept simulations. In this last year of research we refined the SST notional model by including realistic environmental parameters based on experimental data collected during sea trials. Replicating an acoustic environment exactly is impossible – and ultimately not very useful. Instead the goal is to design realistic scenarios in SST so that the behavior of DFH waveforms can be simulated and studied before future experiments.

To more closely approximate realistic environments with SST, first we conducted a detailed study of two eigenray propagation models, GSM and CASS, used by SST. We concluded that CASS is the most appropriate model for our purposes. Then, we refined our previously-developed notional model by converting our previous simulations to the CASS eigenray model and by incorporating realistic environmental parameters based on our measured data from the SPACE08 experiment and on historical environmental data.

Two simultaneous users				
User ID	*Parameter*	*Without eq.*	*With eq.*	*% improvement*
1	Average BER	0.000496	0.000399	19.48
	Median BER	0.000149	0.000075	50
	% of error-free transmissions	0	0	
2	Average BER	0.002072	0.001649	80.72
	Median BER	0.000849	0.000566	33.33
	% of error-free transmissions	0	0	
Two simultaneous users				
User ID	*Parameter*	*Without eq.*	*With eq.*	*% improvement*
1	Average BER	0.003004	0.002810	6.45
	Median BER	0.001610	0.001463	9.09
	% of error-free transmissions	18.20	19.78	
2	Average BER	0.006209	0.005753	54.74
	Median BER	0.004544	0.004251	6.45
	% of error-free transmissions	0	0	
3	Average BER	0.003482	0.003180	8.68
	Median BER	0.001615	0.001468	9.09
	% of error-free transmissions	0	0	
Four simultaneous users				
User ID	*Parameter*	*Without eq.*	*With eq.*	*% improvement*
1	Average BER	0.006372	0.006104	4.21
	Median BER	0.002780	0.002634	5.26
	% of error-free transmissions	12.95	13.51	
2	Average BER	0.011507	0.010974	46.95
	Median BER	0.007476	0.007329	1.96
	% of error-free transmissions	0	0	
3	Average BER	0.043125	0.042678	1.04
	Median BER	0.028194	0.028047	0.52
	% of error-free transmissions	0	0	
4	Average BER	0.017308	0.017170	0.80
	Median BER	0.004267	0.004119	3.45
	% of error-free transmissions	0	0	

Table 2: SPACE08 experiments: performance comparison with and without equalization. There was no single user trial in SPACE08.

Eigenray propagation models

Sound transmission is modeled by SST using eigenrays. Eigenrays are sound rays that propagate from a source at a specific location to a receiver at a specific location. In the context of underwater acoustic communications, ray-based propagation models are independent of the type of modulation[1], so the following discussion applies to virtually all acoustic communication schemes, including DFH. Generally, sound rays starting at a certain position with a specific azimuth (relative to true North) and elevation (relative to the horizontal) undergo changes in elevation and azimuth determined by the sound speed profile (SSP). Furthermore, rays are reflected (and absorbed) by the surface and the bottom. Given a ray from a source, there is no guarantee that it will reach a specific receiver, that is not all propagating rays are eigenrays.

In general, computing eigenrays requires searching a family of propagating test rays to find the rays that arrive at a receiver. It is computationally too expensive to exhaustively search all possible test rays for the eigenrays. More commonly, and with SST, a subset of test rays is searched, and the rays that arrive close to the receiver are interpolated to form eigenrays. For non-trivial SSPs, SST has two ways of computing eignerays: the Generic Sonar Model (GSM) [6] and the Comprehensive Acoustic System Simulation (CASS) [7]. GSM can compute ray traces with either the Multipath Expansion Model (MULTIP), or the Fast Multipath Expansion Model (FAME) [8, 9]. CASS is based on the Gaussian Ray Bundle (GRAB) Model [7].

In the preliminary simulations we conducted at the beginning of this three-year research, we modeled a notional scenario with isovelocity SSP and very simple surface and bottom models. Figure 6 shows the SSP and a set of test rays for this notional environment. The source is at 50 m depth and the receiver is at 55 m depth at a distance of 5 km. The sound speed increases with depth until about 57 m depth. The refracted rays bend upward toward the surface while in that depth interval. In particular, near-horizontal rays will turn upward before they cross 57 m depth, never reaching the bottom. This creates a surface duct for sound propagation. Note that the near-horizontal rays undergo a single surface reflection before reaching ranges of 4.5 km to 8.25 km. Thus, they deliver higher energy at those ranges than multiply-reflected rays, which undergo more severe attenuation. Accurate transmission models must capture these important higher-energy rays, because they account for much of the energy in the peaks of the CIRs, and thus directly affect the multipath characteristics of the channel and the BER of received signals.

During this past year, we investigated the GSM and CASS eigenray models in more detail, and concluded that the CASS-GRAB model is better-suited for more complex and realistic environments, and recommend it for future simulations involving underwater acoustic communications. CASS is better suited for our purposes for several reasons. Figure 7 shows the transmission loss (TL) as a function of range for the GSM-FAME and CASS-GRAB models for the notional scenario. For comparison, a plot of the loss from geometrical spreading is also plotted. First, note that both GSM and CASS predict more loss, that is less transmitted energy, than geometrical spreading. This is expected, because both models more realistically account for ray attenuation due to surface and bottom reflections. More importantly, note that CASS predicts higher energy than GSM at ranges between 4.5 km and 8.25 km, as evidenced by the two bumps in the CASS TL curve. The reason for this difference is that GSM fails to consider the high-energy, single-reflection rays trapped in the surface duct. Similarly, CASS better accounts for the onset of the single-reflection ray at 1.25 km than GSM, as visible in the plot. Accurately accounting for the energy delivered by the rays is of fundamental importance for characterizing the multipath characteristics of the underwater acoustic channel, which directly impact the BER.

Another reason for prefering CASS over GSM is that GSM cannot model range-dependent bathymetry and SSPs, thus limiting its applicability to flat-bottom geographic locations and isovelocity profiles. For these two reasons we argue that CASS is better suited for our simulations.

[1] Except for possible dependencies on the carrier frequency due to environment.

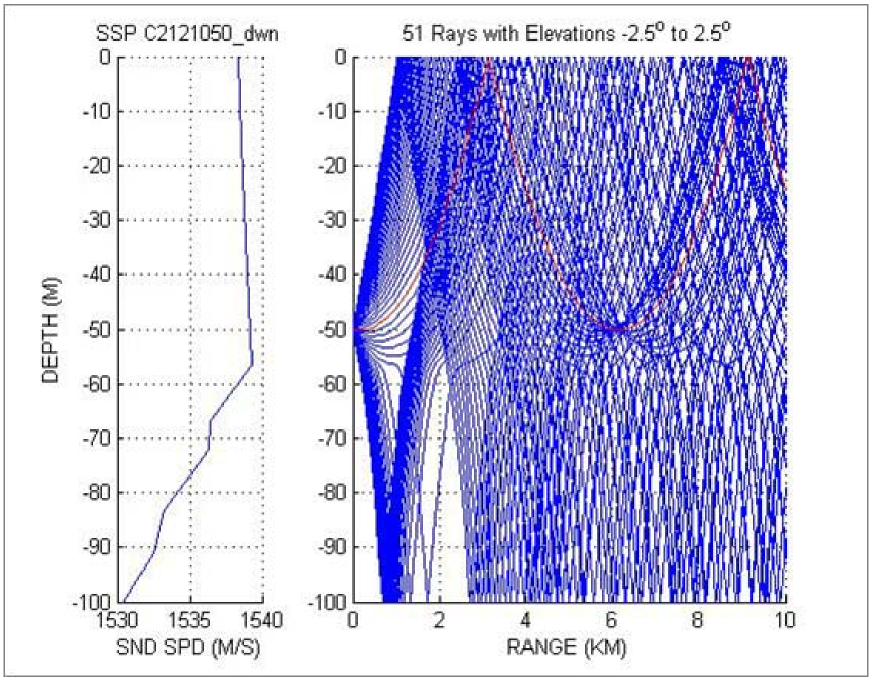


Figure 6: The SSP and resulting rays for the notional environment we modeled. The source is at 50 m depth and the receiver at 55 m depth, 5 km in range. The eigenrays are computed by searching for and interpolating the rays whose path is close to the receiver.

Following our investigation, we converted all our SST simulations to the CASS eigenray models and recommend CASS for future simulations.

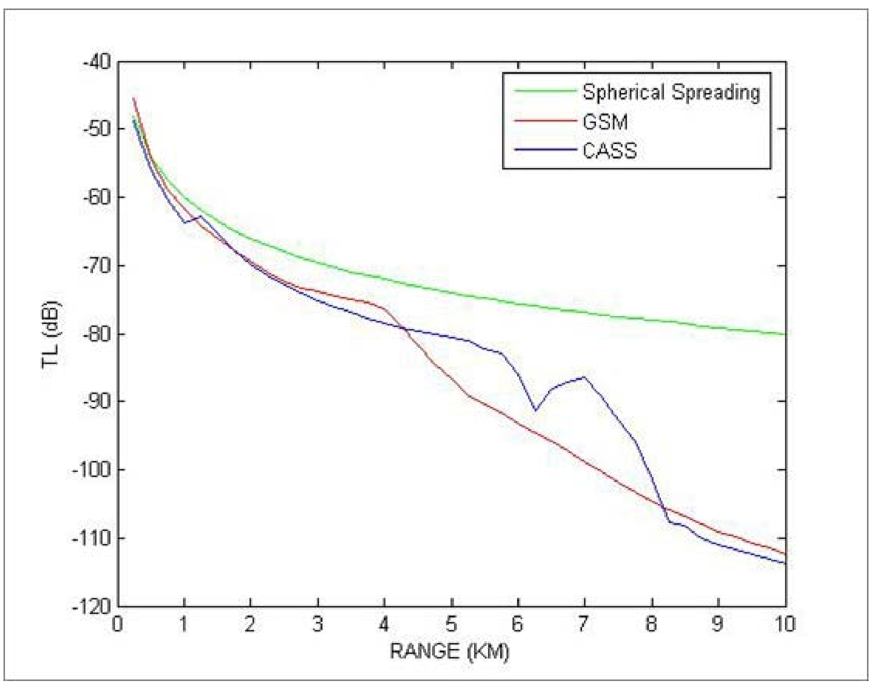


Figure 7: Predicted transmission loss (TL) vs. range for the baseline spherical spreading, and for GSM and CASS eigenray models. Note the bumps between 4.5 km and 8.25 km, which are captured by CASS, but not by GSM, which underestimates the energy at these ranges.

Adopting realistic parameters for SST models

After converting our SST simulations from GSM to CASS, we further refined the models with realistic physical parameters gathered from the SPACE08 experiment. The goal is not to replicate exactly the SPACE08 environment – an impossible goal. Instead, we want to develop proof-of-concept SST scenarios that more closely reflect realistic scenarios for future DFH experiments. Below we detail our progress for various environmental parameters.

Sound speed profile

We estimated a realistic SSP for the SPACE08 experiment as an average of historical data. We extracted all XBTs and CTDs from the 1998 NODC [10] database around Martha's Vineyard (in the square with corners at 21° 45′ N, 159° 56′ W and 22° 6′ N and 159° 28′W) for October and November, computing the corresponding SSP, extrapolating to standard depths, and averaging the results. Figure 8 shows the resulting SSP and test rays. As expected from a winter SSP in northern latitudes, it is basically upward refracting. In fact it is upward refracting at most depths and the degree of refraction is greater than in the previously examined test SSP. There are no direct eigenrays, i.e. all rays have at least one surface interaction. One ray has a single surface interaction and vertexes below the source.

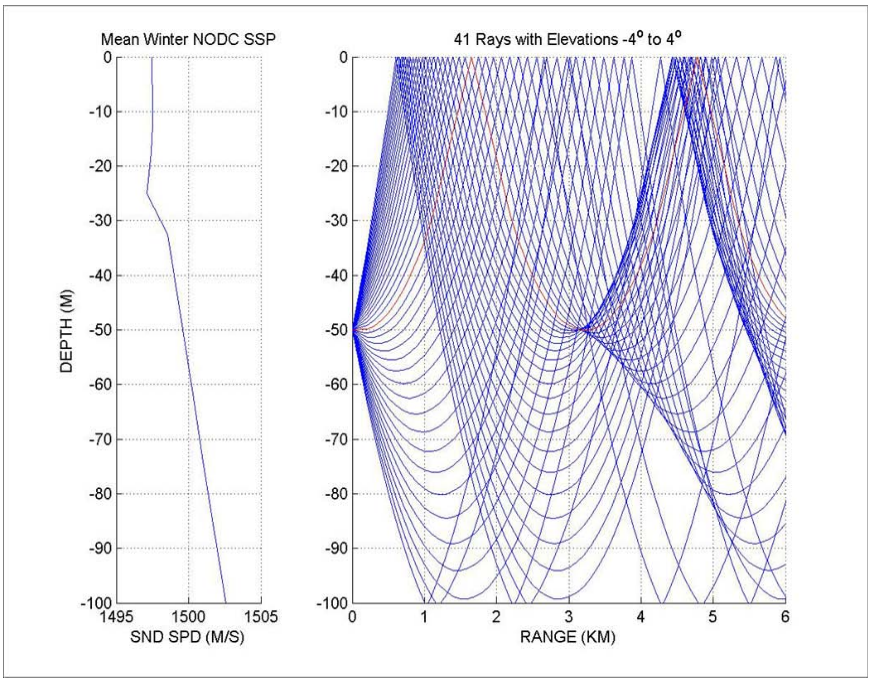


Figure 8: Estimated SSP and corresponding test rays for the SPACE08 experiment.

Bottom reflection loss

The bottom type is sandy [11]. In the 15 kHz frequency range, the standard Navy model for bottom reflection loss is the Jackson model [12]. This model includes interference between layers, but for bottoms without layers such as this one, the main mechanism for bottom loss is sound transmission into the bottom and then absorption. Figure 9 shows the reflection loss as a function of grazing angle for this bottom model. Note that below 30° this bottom model predicts less than 1 dB loss per ray bounce.

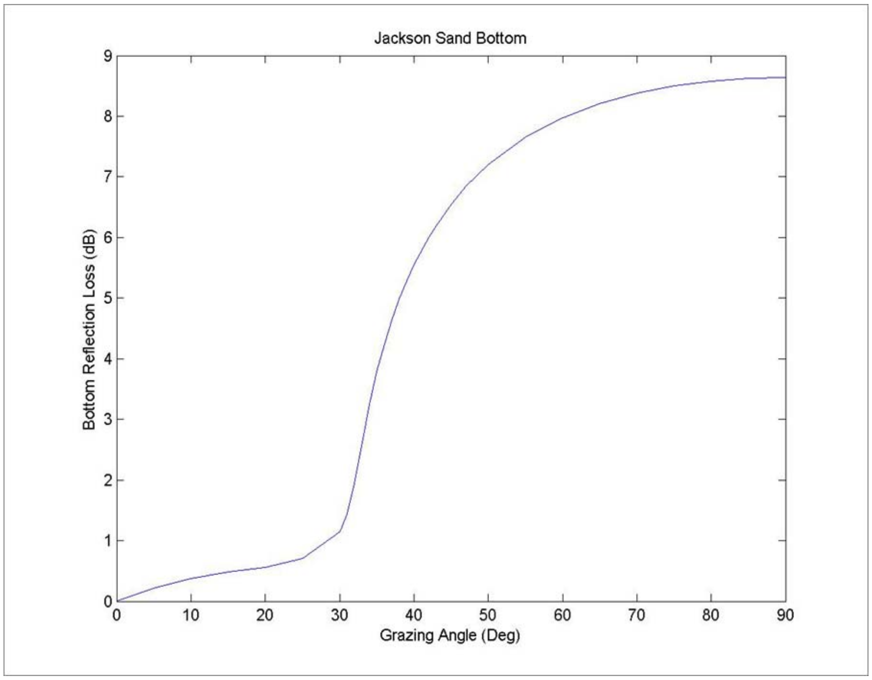


Figure 9: Jackson bottom reflection loss for sand at 15 kHz

Absorption Loss

The absorption loss in the medium was modeled using Thorp [13]. This is not the Navy Standard, but the absorption loss at 15 kHz is about 0.2 dB/km, which is fairly small and so this should represent adequate accuracy.

Surface Reflection Loss

The choice of surface reflection model depends on the type of surface conditions one needs to capture, and no model is universally appropriate for all conditions. Wind speed is one of the most important factors in determining the surface reflection loss, because it directly impacts sound propagation close to the water surface. The increased turbulence due to high wind speeds creates a surface layer of air bubbles mixed with the water. The bubbles absorb sound energy and attenuate the surface-reflected rays, causing attenuation. Figure 10 shows a plot of the wind speed during the duration of the SPACE08 data. Wind speed varied considerably during the experiment: it was generally below 10 m/s in the first week but reached higher levels during the second week. The median wind speed throughout the experiment was 6.3 m/s.

We adopted the APL surface reflection loss model [12], which is the Navy standard for a frequency of 15 kHz. The loss is modeled through a layer of of sound-absorbing bubbles and increases for lower grazing angles, for which the rays spend more time in the bubble layer. Generally, the bubble layer also causes another type of loss, due to a decrease in signal coherence, an important characteristics of the signals we are considering: DFH waveforms are wide-band and we rely on them to remain coherent for error-free transmission. However, for low grazing angles the coherence loss is very small, and the dominant effect on the rays is reflection loss. For this reason we did not include coherence loss in our SST models.

Figure 11 shows a plot of the reflection loss for our setup. Note that below 30°, the loss is very high, which favors high-angle rays with multiple reflections off of the surface and bottom.

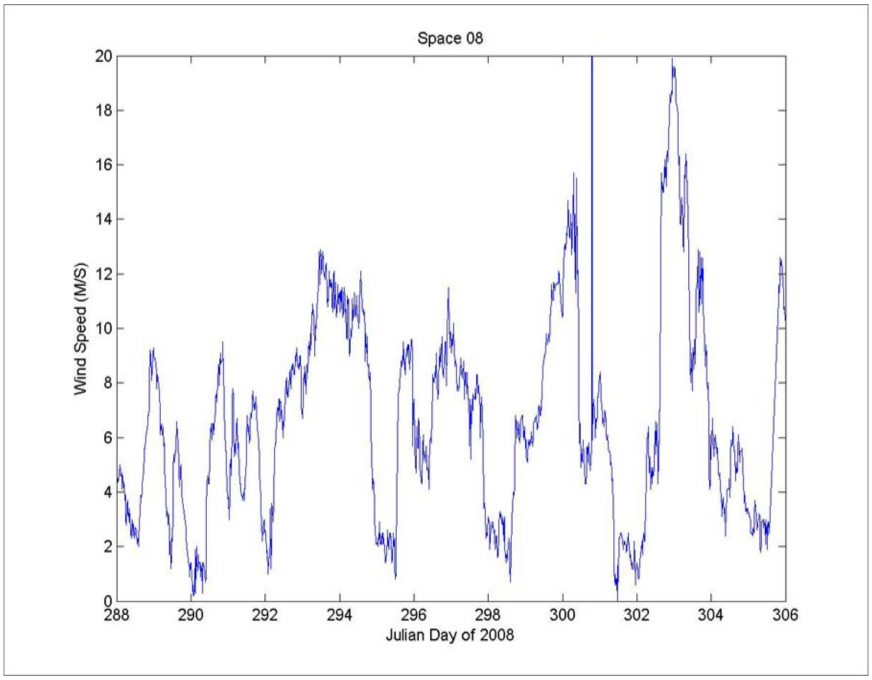


Figure 10: Wind speed at the surface during SPACE08.

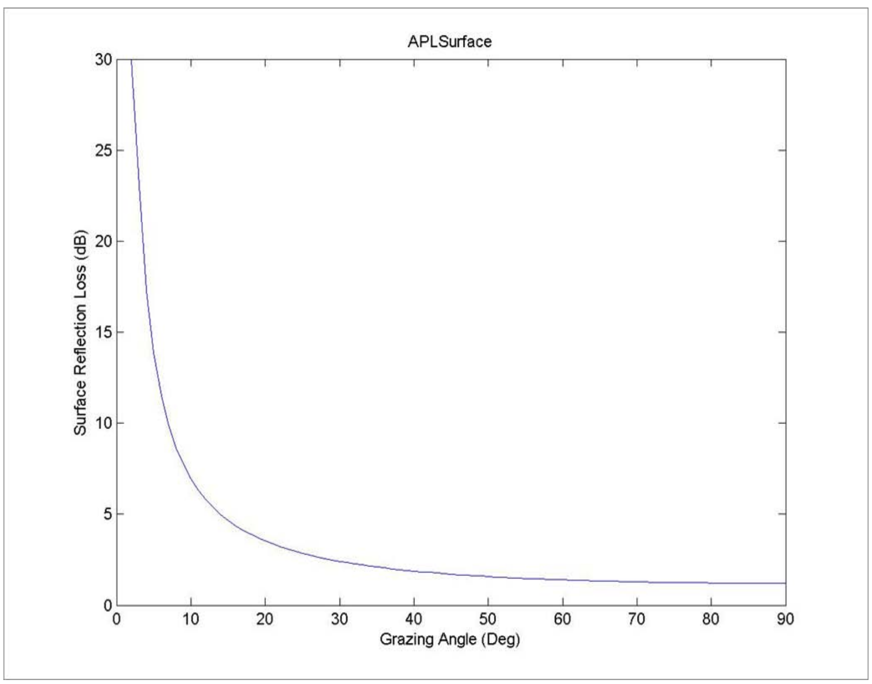


Figure 11: APL surface reflection loss for wind speed 6.3 m/s at 15 KHz.

Channel characterization

We computed the CIR for our refined notional channel to characterize its multipath characteristics. We simulated the transmission of a 1-second LFM chirp probe pulse, for a source at a depth of 50 m and the receiver at a depth of 51 m at 5 km range. The received pulse is replica-correlated with the transmitted pulse to obtain the CIR shown in figure 12. Note the series of peaks: their arrival times correlated well with the eigenray travel times, which we inspected from CASS eigenray file

outputs.

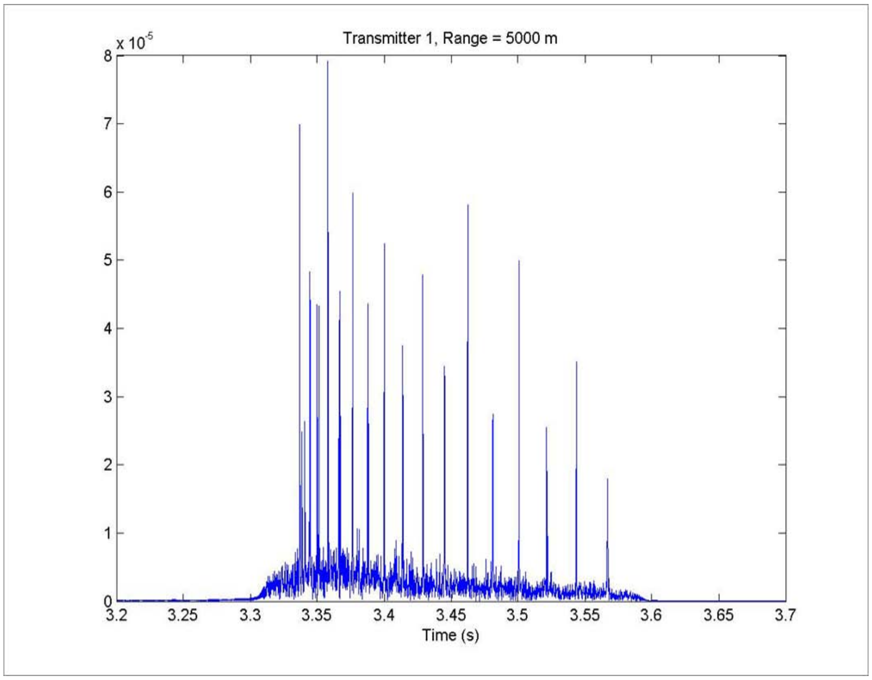


Figure 12: CIR for the refined notional channel, obtained with replica correlation of a 1-second LFM chirp in the 9-12 KHz range.

Publications

We published our results in journals and conference proceedings.

- "Differential Frequency Hopping (DFH) Modulation for Underwater Acoustic Communications and Networking [U]," D. Egnor, G. S. Edelson, L. Cazzanti, and J. Hsieh, U.S. Navy Journal of Underwater Acoustics, April 2009.
- "Underwater Acoustic Single- and Multi-User Differential Frequency Hopping Communications," D. Egnor, L. Cazzanti, J. Hsieh, and G. S. Edelson, Proc. of the OCEANS Conference, Quebec City, October 2008.

Open Questions

The performance of DFH modulation in the underwater acoustic channel has been studied for fixed platforms and for relatively benign environments. Characterizing its behavior for moving platforms and for environment-induced Doppler remains an open area of research. In particular, dealing with both MAI and moving-platform Doppler will require a tight integration of equalization and intelligent demodulation schemes which may include array processing. We conducted preliminary studies which utilized the McDaniel surface model [14] to account for surface motion and observed that the resultant Doppler shift is a considerable effect on the received signals. Robustness to environment-induced Doppler, such as the Doppler spread caused by surface motion remains a challenge.

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